HOW THE SCHOBER REVERBATAPE UNIT WORKS

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There is almost no musical instrument or ensemble which does not sound better in a hall having some reverberation than in an acoustically dead room. This is especially true of an organ — pipe or electronic — since the organ is the most mechanical of all instruments and there is a great deal of "machinery" between the player and his audience (even if the organist himself is his only audience).

Reverberation is the "sound of a large auditorium" which is brought about by the bounces of the original sound against the walls of large, live halls, with consequent audible repititions of each sound at each bounce. The total effect caused by the reverbatory bounces is an apparent slow decay of the sound after each note has ended, rather than the abrupt, cold harshness obtained in a small, padded room where everything is heard with razor-sharp precision and immediacy. Electronic organs, with their scientific precision of sound, are too precise and impersonal to begin with. The defocusing and softening effects of reverberation are really a necessity to give both player and audience the emotional satisfaction and sense of bigness so attractive in a theatre or church organ.

Though not easy to achieve in practice, the requirements for an artificial reverberation system are simply stated. It is only necessary to pass the sound electronically through a device which will repeat each sound at short intervals, each repetition somewhat lower in volume than

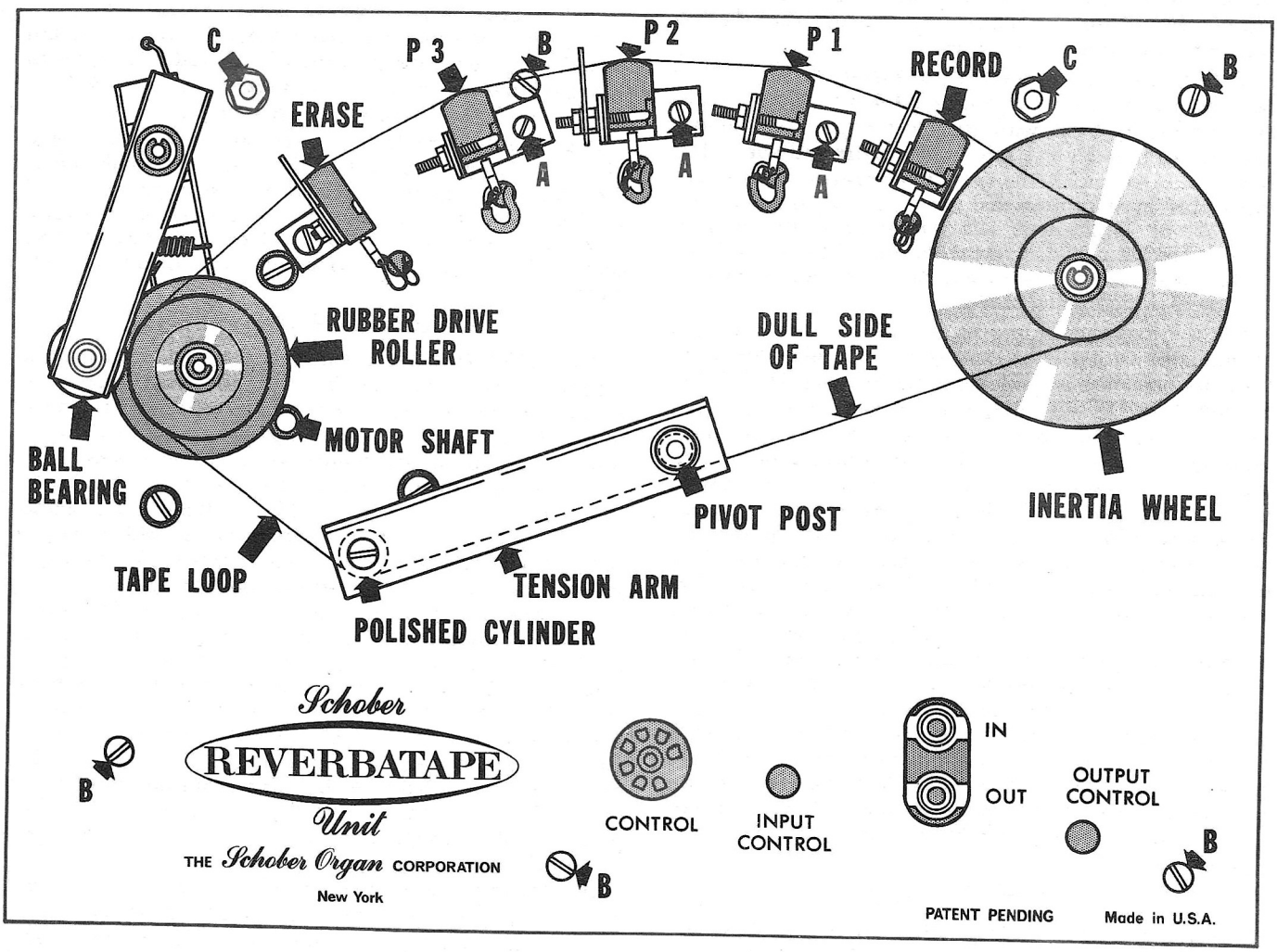

Fig.1 — The tape mechanism of the Schober REVERBATAPE Unit is built on the front face of a steel box. The clear plastic cover, which protects the mechanism and tape, is not shown for clarity.

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the last, until it finally dies away. This is done by the spring-type units which are available today. However, if real satisfaction is wanted, the effect must sound like a natural one; that is, the music should have the same aural and emotional effects as in a real theatre or church, which means in effect that it must fool the hearer, rather than being obvious as just another sound effect. Realism in a reverberator is the ultimate goal, and unfortunately spring devices, which offer the maximum simplicity in design, and the lowest cost, have inherent limitations that make realism impossible. They have sharp resonances, narrow bandwidth-handling abilities (no bass or high treble), and they sound, if anything, like very small rooms with more reverberation than a small room is likely to have, because the period between repetitions is too short. They also cannot be controlled to simulate realistically rooms of various sizes or with walls of varying hardness.

The best method of artificial reverberation is to use a continuous recording device, such as a tape recorder. The sound is recorded on the tape and is played back several times as a given section of tape passes several playback heads. This system has been in use for years in recording and broadcast studios. It has the advantages of complete controllability, in both design and use. It can have full frequency range, low distortion, any timing and reverberation-control characteristics. A good tape reverberator can produce acoustic effects indistinguishable from the real thing. Many phonograph records have been made in small studios, the acoustics enhanced by tape reverberators, without any knowledge on the part of listeners that the job was not done in a large concert hall!

Tape reverberators previously available, however, have been highly expensive — costing $1,000 to $3,000. They are customarily built with the most expensive components, are completely variable both mechanically and electronically, and employ the type of construction one would expect for hard professional use in a very large variety of different applications. An important factor in their pricing has been that so few were sold (how many large professional users of such equipment can there be?) that they were almost hand-made.

For the first time, however, a relatively low-cost tape reverberator with all the quality and durablity necessary for daily home use is available in the Schober REVERBATAPE* Unit. It was designed especially for electronic organs (though it is equally usable in many other applications) and does not have the wide adjustabliity of the expensive professional units. Its mechanical system is simpler than that of the least expensive tape recorder; yet because it has only one job to do — no stop, reverse, fast or slow, record, erase, or playback switching — it is as reliable as the good conventional professional recorders and has low wow and flutter content. Electronic circuitry, which is fully transistorized, is relatively simple, with no large power supply or power amplifier, and cost is further reduced by quantity production. It is still considerably more expensive than many of the spring reverberators, but its audible results are so realistic that the user quickly forgets that it is present — until he happens to turn it off! Professional musicians have commented that it is the one advance in the electronic organ art that is likely to make the electronic organ acceptable to the most reactionary pipe organist.

A Bird's-Eye View

In general, the REVERBATAPE Unit operates in pretty much the classical manner of devices of this kind, which date, at least in concept, back to the beginnings of tape recording; though in specific details many aspects of the Unit are quite new.

There is a continuous loop of tape which continuously passes through the machine, running in turn over a recording head, three playback heads, and an erase head. Program material, such as the output of an electronic organ or a musical performance to which reverberation (large auditorium) effect is to be added for a recording, is fed to the input of the Unit.

The output of the Unit, which is fed to the usual power amplifier and speaker (or, depending on the use, to the recording machinery or broadcast circuits) contains first the original program material, merely passed through the REVERBATAPE Unit without change of any kind. Second, and more important, it contains program material which has been recorded on the tape, then picked up by each of the three playback heads in turn. Since the tape takes time to get from the record head to each playback head (about 0.11 second), each playback-head signal is a delayed repetition of the original material.

The third playback head, in addition to feeding the third delayed repetition to the output, also feeds a signal back to the record head, so that the process is repeated for another tape pass across the heads. The result is a series of equally time-spaced repetitions, each repetition decreasing by a calibrated amount from the previous one until finally the signal is too small to be heard. The effect is exactly that of an auditorium where repetitions are obtained by bounces from the walls.

The user controls the effective length of the decay, and thus the apparent characteristics of the "auditorium," with a single knob which varies reverberation time (time for a 60-db decay) from zero to over 6 seconds.

A great many problems were encountered in the design of such a reverberator, all of them linked: lifelike effect (the first requirement), reliability, cost, tape wear, and the like. How these problems have been overcome over a 4-year development period to produce a reliable, effective commercial device is described in some detail in these pages.

REVERBATAPE Mechanics

The REVERBATAPE Unit consists principally of a tape-handling mechanism built on the front face of a heavy steel box, shown in Fig. 1. The tape itself is a very special kind of Mylar-base computer tape. The oxide is dispersed in an unusual binder which has enormous holding power, so that continuous passing over magnetic heads causes very little deterioration. The small amount of oxide rubbed off comes away as a dry powder, which simply falls, rather than as "gunk" which remains on the head gaps and fouls them, as with the usual tape.

The tape is in the form of a single continuous loop only 19 inches long. In addition to the loop already on the machine two spares are supplied, each of which is guaranteed for 500 hours of operation. In laboratory tests single loops have been run for over 1000 hours without audible degradation. In 500 hours of operation the tape passes over the magnetic heads 1,137,081 times, which, because the REVERBATAPE Unit has 5 heads, is an even greater beating than one could give a recorded tape by playing it this same number of times on a standard tape recorder!

Besides excellent wear characteristics, such a short loop of tape must have one other very important property — a strong and virtually inaudible splice. Since the splice passes the magnetic heads once every second and a half, it is obvious that any dropout, noise, or speed variation

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due to the passage of the splice over the heads could be very annoying. A standard tape splice, made with presure-sensitive splicing tape, is unsatisfactory for two reasons. First, such a splice slowly pulls apart due to the constant tension applied to the tape during many hours of continuous use. This results in an audible dropout and speed variation at first and, finally, separation of the ends of the tape. Secondly, a standard splice stiffens the tape at the splice point and creates a discontinuity in the smooth flow of the tape through the machine.

The problem of providing a piece of tape free of discontinuities cannot be solved by simply using a continuous oxide-coated loop of Mylar; no such an animal is available! However, it is possible actually to *weld* the tape together with heat and pressure, and that results in a very close approximation to the ideal of the truly continuous loop. With the welding process, the splice point is nearly invisible on the oxide face side of the tape. The oxide is perfectly smooth and continuous and the operation of the tape, mechanically and electrically, is as if the tape were one continuous piece.

The tape used in the Schober REVERBATAPE Unit is formed into loops by the welding process using a specially built (and rather expensive) machine. This machine produces a loop of exactly the right length and makes the weld splice at a 45-degree angle.

Tape Drive

The tape is driven by a small 4-pole motor within the box, whose steel shaft emerges as shown in Fig. 1. The rim of a rubber drive roller on an arm is very lightly pressed against the motor shaft by a wire spring, and the tape rides over another section of the roller. For positive drive, the tape is pressed against the roller by a ball bearing at the end of another arm under spring pressure.

When the machine is not running, the pressure of the rubber roller rim against the motor shaft is light enough so that no permanent indentation is made in the rubber by the shaft. However, when the machine starts, the fact that the roller is pulling the tape against some resistance tends to snub the roller against the motor shaft. The roller, in effect, pulls *itself* up tight against the shaft. Also, as the motor shaft drives the rubber roller, a force is created which tends to swing the arm on which the roller is mounted *toward* the motor shaft, further pressing the roller and shaft in firm contact. This is a "self-energizing" system, unique in tape mechanisms to our knowledge, which does away with any need for mechanical disengagement of the two when the Unit is shut off, and considerably simplifies the system. The tension arm is the rectangular part near the center of the mechanism which is pivoted at one end and has a highly polished cylindrical surface at the free end. The tape passes under this arm and is in contact with the cylindrical surface. The weight of the arm pressing down on the tape has been designed to provide just the right tension to hold the tape in firm contact with the magnetic heads without pressure pads. Since the tape tension is controlled by gravity instead of springs, it can never vary or get out of adjustment so long as the Unit is operated right-side up on a planet with Earth gravity. After the tape passes beneath the tension arm, it rides over the inertia wheel which is located on the upper right-hand corner of the mechanism. This precisely machined brass wheel runs free on its shaft; its function is to smooth out any variations in speed. Its counterpart is the flutter filter seen on many tape recorders.

Head Spacing

The tape runs over the record head, three playback heads, and an erase head, in that order, at a speed of about 12 inches per second. The separation between heads is approximately uniform, and is such that a given point on the tape takes about 110 milliseconds (0.11-second) to go from one head to the next. After passing over the rubber drive roller, tension arm, and inertia wheel, a point on the tape again passes over the record head to be recorded again.

What about the selection of 110 milliseconds as the period between reverberatory repetitions? That may seem a little long from one standpoint. Many workers have felt that it is a prime necessity to keep the period between repetitions short enough so that under *no* circumstances can repetitions be heard separately rather than blending into an apparently continuous decay. This approach has some merit, and in a real auditorium there are so many different reverberation periods due to the many different trains of repetitions from various surfaces that discrete repetitions are almost never apparent.

It is an empirically proven fact, however, that even in the typical auditorium, which has so many complex reverberation paths, the ear seems to "select" the longest period between repetitions, which represents the reflecting surfaces which are farthest apart, and judges *from that alone* the size of the auditorium. And in an artificial system, if all the periods between repetitions are short the hearer gets an impression of a small room — which is unpleasing to begin with and may also sound highly unnatural if the system can at the same time give long decay times which would never occur in such a small room. This is one of the main failings of spring systems.

It is therefore far better to choose the longest period between repetitions which can be tolerated without causing discrete and separate sounds *in most cases,* because this sounds like a large room. A note must last at least 110 ms to avoid discrete repetitions in the REVERBATAPE system, but this is still a very short staccato, and the repetitions will be apparent only near the full-reverberation control settings. Faster music, which is likely to have larger numbers of staccato notes, would always call for a reduced reverberation-time setting to attain clarity. Then any discrete repetitions that may occur do not last long and are covered or masked by following music. The compromise of 110 ms has proved over a long period and with the widest possible variety of music to be a good one.

It can, however, cause difficulty with some kinds of program material if improperly used. The sounds of speech are extremely staccato, and so are those of some instruments such as piano and harpsichord. When it is used for these sounds, the control must not be turned up too high. Oddy enough, when the Unit is used with a guitar amplifier, no problems arise. Evidently, the decay of sound after a guitar string is plucked is so smooth that it approximately follows the reverberative decay.

Electronic Requirements

The initial specifications upon which the total design is based were arrived at on a purely subjective basis. While the science of acoustics can tell us — at least in part — how to achieve certain reverberation characteristics in building a hall, the question of what these characteristics should be is still a matter of what an accomplished and experienced musician feels is best. This subjective approach is entirely valid because music itself, being an art, is intended to affect the emotions. Man has not yet learned to analyze himself enough to make reliable correlations between scientific criteria, such as meter readings, and human emotional reactions.

The equipment used for determining the design object-

ives was a large, specially built machine mounted in a 6-foot rack. It had a large tape deck with means for driving tape at various speeds and several record-playback heads which could physically be moved to obtain any desired separation between them. The rack had 6 amplifiers, a multitude of switches, and controls for volume and equalization, and facilities for doing almost anything. In several months of experimenting with number of heads, head placements, tape speeds, equalization to determine relative reverberation times for various parts of the audio spectrum, and the like, the final requirements for the production units were obtained. Except to determine what had been achieved finally, meters were not used; for every test, organ and other music was actually played through the system and listened to by several sets of trained ears. Only two results were desired: the utmost realism; and the ability to control reverberation to fit various kinds of music while maintaining complete realism at each control setting.

It was discovered during this phase that the control methods usually used in artificial reverberation systems — especially the spring units — was unrealistic. Usually there is a direct channel, through which sound goes without added effect, and a parallel reverberation channel, in which reverberative repetitions are added to the original. These are combined at their outputs. To reduce reverberation time, a control simply reduces the gain of the reverberation channel, so that relatively more direct sound is heard.

Fig. 2 shows the result. With full reverberation setting, the direct sound is represented by line A. The sound source ceases (the player releases a key) at time X. Then the reverberation channel causes a slow decay along line B.

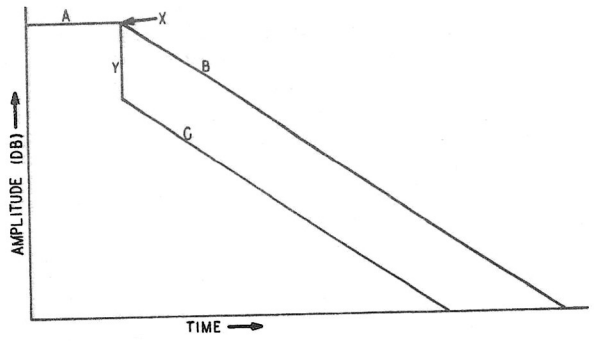


Fig. 2 — The unnatural effect of controlling a reverberator by simply reducing the relative gain of the reverberation channel. Abrupt drop in level occurs after sound source ceases, followed by decay with unchanged slope.

When the reverberation channel is turned down, the direct sound A stops at time X. There is then an abrupt drop in level (Y), after which the decay occurs at the same rate (C) as before. No real auditorium does this. Instead, the decay always begins smoothly just when the sound source is removed. Fig. 3 shows this effect at various total reverberation times. Each decay curve B, C, D, E, is a smooth one beginning exactly at point X, without any abrupt drop after cessation of source. The decay slopes are, of course, different. To accomplish this necessary bit of realism in the REVERBATAPE Unit a new kind of attenuation system had to be designed — one which would place the output level at each reverberator repetition a certain number of decibels less than the last at all control settings, the certain number of decibels changing with control settings. This is described in detail later.

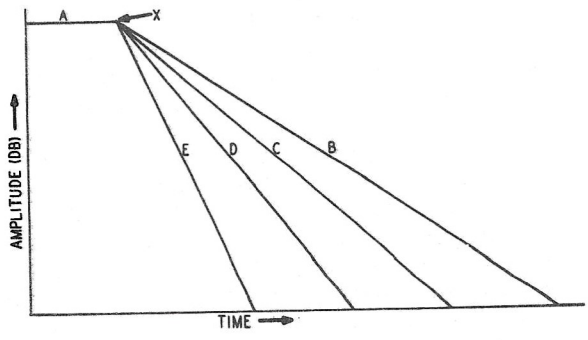


Fig. 3 — Natural decay characteristics for various reverberation times, as given by the REVERBATAPE Unit.

Circuitry

Being essentially a tape recorder, though a very special one, the Unit requires the usual stages — record amplifier, bias-erase oscillator, a playback amplifier for each head — plus special stages. The electronic section employs 11 identical small-signal transistors, two power transistors, and a zener diode and four silicon diodes in the power supply. All the circuitry is built on a single printed circuit which may be seen in the rear-view photo of Fig. 4. The

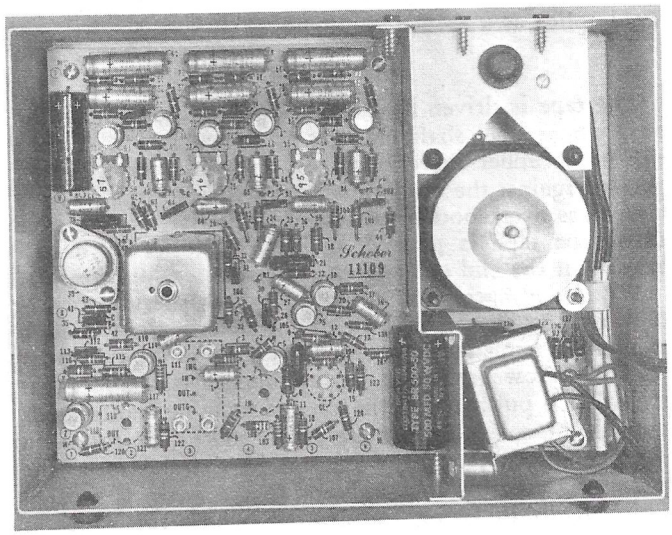


Fig. 4 — Looking at rear of REVERBATAPE Unit, the printed circuit and motor can be seen.

board even contains the power supply (except for the power transformer), which furnishes only about 50 ma at 30 volts, about 1.5 watts of power for the whole electronics! The supply is regulated. The fully enclosed power transformer can be seen at the lower right in Fig. 4. The mounting of the transformer at an angle may seem a bit odd, but this is a condition for very minimum hum.

Fig. 5 is a complete schematic diagram of the RV-3 REVERBATAPE Unit. Everything shown is on the printed circuit except the motor, fuse, and power transformer.

The Unit is usually operated with unity gain. A presetting system makes it possible to adjust the Unit, as normally supplied, to obtain maximum signal-to-noise ratio (maximum record level on the tape) as well as unity gain, over a range of maximum signal level of .3 to 3 volts rms.

The presetting system consists simply of independent input and output controls, each having a total range of approximately 20 db. The input control is set for whatever maximum signal level is available within this range and the output control is adjusted so that the same signal level appears at the output. Therefore, as long as the

maximum signal feeding the input of the Unit falls within the .3-to-3-volt range, the Unit can be adjusted for maximum signal-to-noise ratio while maintaining a one-to-one signal ratio between input and output.

Adjustment for unity gain is not actually necessary. There may be either a gain or loss between input and output, as seems desirable in a particular application, since input and output controls are adjusted separately. This means that the Unit may have as much as 20 db gain (input set for 0.3 volt, output set to yield 3 volts as a result) or as much as 20 db loss (input set for 3 volts, output set to yield 0.3 volt).

The input stage, transistor 5, has a gain of about 26 db from base to collector. The resistor 2 (470,000 ohms), in series with the input, provides a loss of about 26 db of the input signal. This is due to the voltage divider composed of this resistor and the base-to-ground impedance of about 22,000 ohms. It can be seen, then, that the input impedance of the Unit, as normally supplied, is nearly 500,000 ohms and the net gain for the first stage is 1. This high impedance makes the unit suitable for insertion into even high-impedance vacuum-tube circuitry without running into loading problems.

As one would suspect, it is perfectly feasible to reduce the value of resistor 2 to increase the sensitivity of the Unit, if desirable or necessary in some applications. Of course, the input impedance will decrease as the value of 2 is reduced. The maximum sensitivity possible is .015 volts and is achieved when 2 is zero. Under these conditions the input impedance is about 22,000 ohms.

The input stage is of the "H" configuration having most of the emitter resistance bypassed (11) sufficiently to keep the response flat to well below the audible range. Part of the collector resistance, 6-7 is the input control 7.

The 1,000-ohm part (8) of the emitter resistance is bypassed by 9 which is small enough to result in a response at control 7 which begins to rise at 2 kc and reaches +12 db at 10 kc. The response at this point is shown by curve A of Fig. 6. This preemphasis (which is cancelled in a later stage by a complementary deemphasis) is instrumental in reducing to a very low level the hiss at the output of the Unit.

The signal leaving the input stage is applied simultaneously to the output stages and to the recording system. The signal passed to the output stages eventually emerges at the output unaltered in any way. This is called the *direct signal* and is the original sound of the program source.

The signal going to the recording channel is first passed through a voltage divider (13 and 14) before going through control A (one section of the three-gang REVERBATAPE Control) and finally reaching the base of transistor 19, the recording preamplifier. The purpose of the voltage divider is two-fold: it adjusts the signal level to that required at the base of transistor 19 and provides the proper source impedance for control A. Capacitors 12 and 16 are simply d.c. blocking capacitors.

The recording preamplifier stage, transistor 19, is virtually identical to the input stage. Capacitor 21 provides preemphasis as in the input stage but for a different reason.

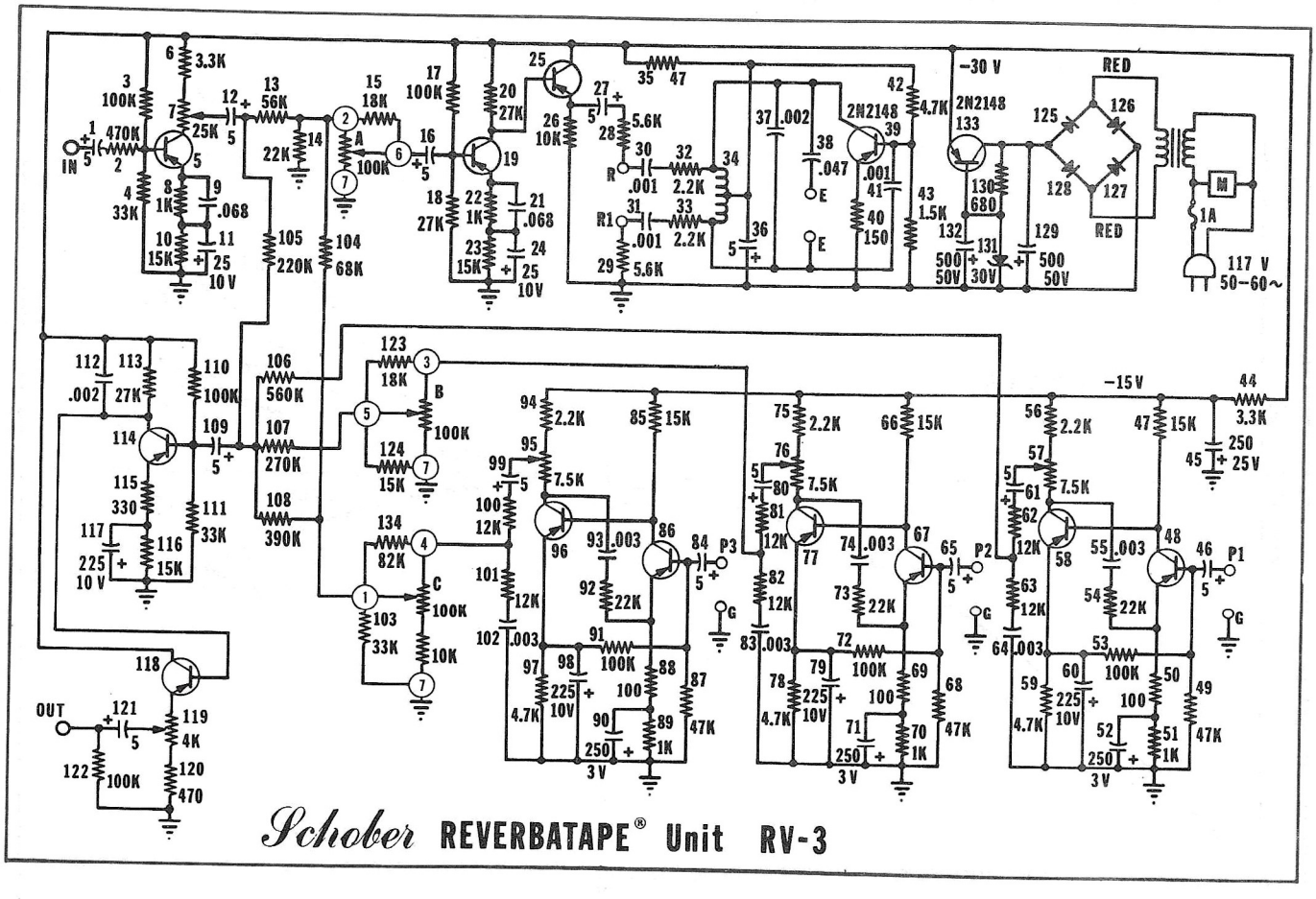


Fig. 5 — Complete schematic of the REVERBATAPE Unit RV-3. The 12-transistor circuit plus power supply (except for the transformer) is all on a printed-circuit panel, which is plainly marked, front and back, with identifications of component locations. Only the Control (ganged potentiometer A, B, and C) is external to the Unit.

Here, treble preequalization is introduced to compensate for the high-frequency playback loss of the magnetic playback heads. The response at the collector of transistor 19 is that shown by curve B of Fig. 6.

The output of transistor 19 is direct-coupled to the base of transistor 25, which is the recording amplifier. This stage operates as an emitter follower with the record head connected from emitter to ground through suitable series resistance. This resistance is divided into two equal parts, 28 and 29, with the record head inserted at terminals R and R1. Capacitor 27 blocks the emitter d.c. voltage. The series resistance totals about 11,000 ohms, which is sufficient to cause constant audio current at all frequencies up to about 10 kc, dropping off to —3 db at 20 kc relative to the 1-kc level.

Bias and Erase

The bias-erase oscillator is a single-ended circuit using power transistor 39 to provide sufficient current with low harmonic content. As is well known, a good bias sine wave produces minimum tape noise.

There are two unusual points about this oscillator. First, the erase head (which is connected between the E terminals) is part of the inductance which, with capacitor 37, forms the resonant circuit which controls the oscillator frequency. (Capacitor 38 blocks d.c. and is relatively large.) The Q of this resonance is very high, which, in addition to stabilizing the oscillator frequency, tends to keep the waveform clean. Part of the high current circulating in the resonant circuit passes through the erase head and that part is just what is required for good erase.

The bias voltage to be applied to the record head is available across the oscillator coil 34, which is also part of the tank inductance. The record head is connected across the coil through resistors 32 and 33 and capacitors 30 and 31 as shown in the schematic diagram. The resistors adjust the bias voltage and the capacitors block the d.c. The bias-erase frequency is between 55 and 60 kc.

Playback Amplifiers

The three playback heads are connected to points P1, P2, and P3, the inputs to the three playback amplifiers. Since the three amplifiers are identical in every respect, only one need be discussed.

Consider the amplifier for the first playback head, which consists of transistors 48 and 58. The amplifier incorporates d.c. and a.c. negative feedback. The equalization necessary for the standard correction of the head playback characteristic is incorporated in the a.c. feedback connection and is composed of 54 and 55.

The well known advantages of distortion reduction and increased input impedance are, of course, also gained. As a result, the base impedance of transistor 48 is about 20,000 ohms above 1 kc. With an inductance of 90 mh for the playback heads used, there is no high-frequency loss (due to the low-pass action of the head inductance and the 20,000-ohm base impedance) below 20 kc. The coupling capacitor 46 is large enough to allow the bass response to be flat to below 20 cps.

The frequency response of the signal picked up from the tape by the playback head, ready for equalization, is that of curve C in Fig. 6. This is the signal that enters the input of the playback amplifier at P1. Twelve db of the total 10-kc preemphasis has been eliminated by the high-

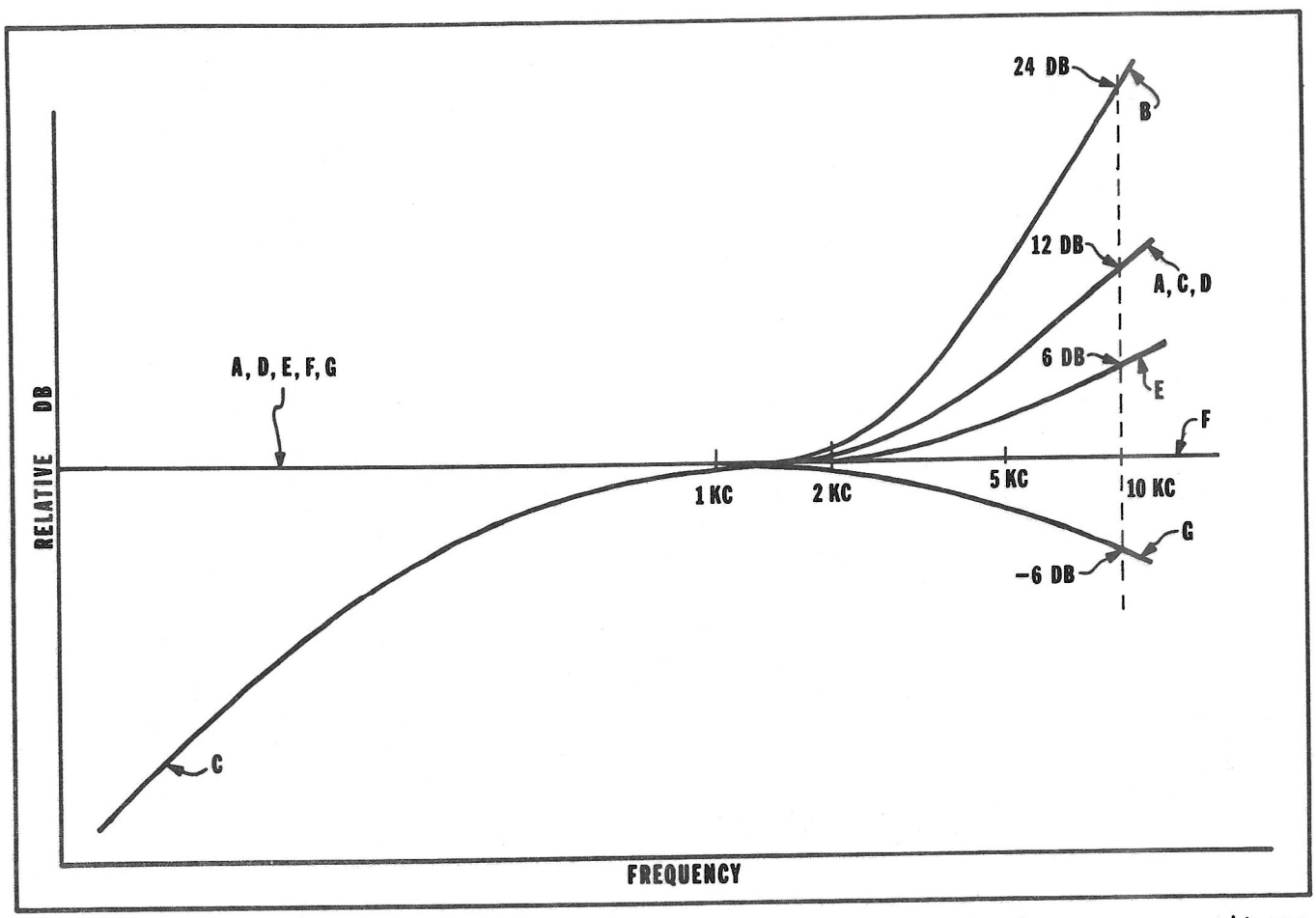


Fig. 6 — Frequency response at various points in the REVERBATAPE circuitry. The text explains why these various response curves exist, even though the over-all response of the Unit is flat.

frequency playback loss of the head. The remaining pre-emphasis is due to that built into the input stage and is removed in a later stage. The bass attenuation (curve C) follows the normal result for pickup from a tape recorded with constant audio current in the record head.

The feedback equalization corrects this by producing a complementary bass boost which results, at the collector of transistor 58, in curve D of Fig. 6.

The d.c. negative feedback is obtained by connecting resistor 53 from the emitter of transistor 58 to the base of transistor 48 and direct-coupling of the two stages. Fifty-three and 49 form a voltage divider which adjusts the emitter voltage of transistor 58 to that required at the base of transistor 48. The bypassing of 59 by 60, besides eliminating all emitter degeneration in this stage, prevents a.c. feedback from occurring in the d.c. loop.

The d.c. negative feedback method of applying the necessary biases has much to commend it. First, biasing, with good temperature stability, is accomplished with fewer resistors and a coupling capacitor is not required between the two stages (in fact, must not be used, or the d.c. feedback loop will be broken). Second, the amplifier as a whole is set up as one integral unit. Such factors affecting bias as temperature and d.c. amplification can change in one stage and be corrected in the other.

Output from the amplifier is taken from potentiometer 57, which, with 56, forms the collector resistor for transistor 58. The potentiometer makes it possible to adjust the amplifier output to accommodate the normal production tolerances in magnetic head sensitivities. This adjustment is made as a factory calibration. The blocking capacitor 61 prevents d.c. from being applied to the following sections of the three-gang reverberation control.

High-frequency rolloff must be incorporated in the playback channels in order to provide a more rapid decay of high frequencies compared to the low frequencies in the reverberated signals. In a building, highs are absorbed more rapidly than lows due to selective attenuation in the air and at the reflecting surfaces. Resistor 63 and capacitor 64 accomplish this objective by producing a very slow rolloff amounting to about 6 db at 10 kc. At this point the frequency response is that shown by curve E in Fig. 6.

Mixing and Output

All three playback amplifiers are identical. The output signal of the first is passed directly to the mixer, transistor 114, through resistor 106. The outputs of the second and third playback amplifiers are passed through reverberation control sections B and C respectively, then through resistors 107 and 108 respectively, and finally to the mixer stage. Resistors 106, 107, and 108 prevent loading and resultant upset of the control tapers, which are critical, as well as isolation. The direct signal is also mixed into this stage through 105.

The mixer stage (transistor 114) is basically the same as the input stage with the exception that high-frequency *deemphasis* takes place here. The base of the transistor is presented with the direct signal which, it will be recalled, has the frequency response of curve A in Fig. 6 after leaving the input stage. The play back signals, also appearing at the base of this stage, have the frequency response of curve E in Fig. 6. High-frequency deemphasis is accomplished by 112 which shunts the collector resistor at high frequencies, causing a rolloff which begins at 2 kc and becomes 12 db down at 10 kc. The output of the mixer, therefore, provides the net response of curve F for the direct channel and a response of curve G for the playback channels. Note that the direct signal is now flat, unaltered compared to the original input.

The response shown in curve G for the playback channels exists for only the first 0.33 second after the input signal stops (such as at rests or at the end of the music). After this point the signal heard is a "secondary" recording, picked up from the third playback channel, passed through 68, and recorded again. When this signal emerges through the playback amplifiers, the high frequency response is further rolled-off because an already rolled-off signal was used to record it! In this way the high-frequency response of the reverberation signals is steadily degraded as the decay progresses. This exactly duplicates the way natural reverberation occurs in a well designed auditorium and is an important final touch to the realism with which the REVERBATAPE performs.

The output stage of the Unit is emitter follower 118 which, of course, supplies the well known advantages of a low-impedance output. This stage is directly coupled from the mixer stage, its stability being controlled by that of the well stabilized mixer stage. The emitter resistance is composed of 120 and the output control 119, which are proportioned to provide a 20 db range of the control. The output impedance of the Unit depends on the setting of the control, varying from about 50 ohms to about 1000 ohms, low enough for all normal applications.

System Noise

As is true of any good tape-recorder design, saturation of the magnetic recording tape in the REVERBATAPE Unit is the controlling factor in just how hard the input can be driven. Maximum signal-to-noise ratio results only when the unit receives the highest possible input signal short of tape distortion. Therefore, any overload of the electronics that prevents the fullest possible recording on the tape can only result in a lower signal-to-noise ratio.

In the REVERBATAPE Unit the minimum reserve between overload of the tape and overload of the electronics is 6 db for any combination of input and output control settings, and it occurs in the playback amplifiers. This means that when the tape is recorded to its limit, the electronics are operating well below the distortion point. The maximum reserve occurs in the record preamplifier and amplifier — a hefty 17 db.

In spite of the fact that the REVERBATAPE Unit has the peculiar characteristic of having the noise output of three tape recorders, but the net audio signal of only one, plus rerecording feedback, the signal-to-noise ratio is about 50 db below maximum undistorted output at *maximum reverberation,* very comparable to the ratio specified for high-quality broadcast tape recorders. At the more normal setting number 7 on the Control dial, the ratio is about —53 db; with reverberation off, about —55 db. These readings include all audible hum and noise below 20 kc.

Control System

The control system is one of the most important REVERBATAPE features. (A patent is pending.) The pipe organist in a church is stuck with whatever reverberation characteristics the church happens to have, but no one characteristic can suit all kinds of music. Reverberation time is defined as the time required for a tone of 512 cycles per second (an octave above middle C) to decay 60 decibels. If this takes perhaps 6 seconds, the hearer receives an aural impression of a very large and impressive auditorium. However, such a time is suitable only for very slow, stately music. Faster music becomes blurred and unclear because while any notes are being played the sounds of the notes played almost 6 seconds ago are still present. An extreme and very well known case is the Cathedral of St. John the Divine in New York, where the reverberation

time is around 9 seconds. While a reverberation time this long is most impressive for very, very slow music, it makes the playing of faster, more detailed and complex music, such as counterpoint, very difficult, and the aural result is muddy and unclear. Nonetheless, long reverberation times are sometimes desired for novel effects.

The REVERBATAPE Unit is so calibrated that when the control is at maximum the output of the Unit due to signal from the first playback head is about 1 db less than the output due to the direct signal. Output due to the second head is 1 db less than that, and signal due to the third head is still another decibel lower. Output of the third head is also fed back to the record amplifier and re-recorded, so that this cycle continues — each succeeding repetition 1 db less than the last.

With this control setting, and with the repetitions spaced 0.11 second apart, it takes 60 repetitions or 6.6 seconds for an attenuation of 60 db, and this is the maximum reverberation time of the device. In testing, the maximum reverberation time of the unit may not seem this long unless the original sound is loud enough so that after 6 seconds the sound, which is now down 60 decibels, can still be heard! The ear actually senses the *rate* at which the sound is decaying.

Faster music, or music requiring greater clarity, however, requires shorter reverberation times. These are obtained when the control is rotated counterclockwise by increasing the attenuation between played-back repetitions. For example, when the setting gives 2 db between repetitions, only 30 repetitions or 3.3 seconds, is needed to reduce level by 60 db. The control is continuous from maximum down to zero. The following explanation of how it works requires careful attention to avoid confusion.

Maximum reverberation time is obtained with the sliders of controls A, B, and C all the way up. (These controls are ganged on one shaft and make up the REVERBATAPE Control, which is available to the organist.) Control A then sends full signal to the recording amplifier 25. Signals from the three playback amplifiers join at the base of transistor 114. With the Control all the way up, the calibration potentiometers 57, 76, and 95 have been so set at the factory that the signal reaching 114 due to head 2 is 1 db less than that due to head 1; and the signal at 114 due to head 3 is 1 db less than that due to head 2. In order words, the outputs due to the three playback heads are 1 db apart.

Let us specify certain terminology. Let us always refer arbitrarily to the level of signal appearing at the Unit's output terminals *due to passage through the direct channel and not from the recording system* as 0 db. With the Control at maximum, the three calibration potentiometers 57, 76, and 95 have been adjusted so that signal appearing at the output terminals due to playback head 1 is at a level of —1 db, that due to head 2 at —2 db, and that due to head 3 at —3 db.

Signal from playback head 3 not only goes through resistor 108 to join the other signals at the base of 114; it also goes through resistor 104 to control A for rerecording or feedback. This signal appears at the control along with the signal from the Unit's input. However, the signal appearing at the control from head 3 is 3 db lower in level than the signal due to the device input. This 3-db-less relationship, which is determined by choice of the components, is always present. Since the signal obtained at the base of 5, recorded and played back through head 1, resulted in a system output of —1 db, the fed-back signal produces a system output 3 db less than that, or —4 db. This is 1 db less than the output due to the original head 3 signal, and it continues the scheme of continual 1-db-apart repetitions.

We may also consider that, due to the setting of calibration potentiometer 76, the output of head 2 has a permanent 2-db attenuation, and that of head 3, due to the setting of 95, has a permanent 3-db attenuation. Adding 3 db to each of these for the second round of signals produced as the result of the feedback gives 3 db less than the original —2 for head 2, or —5, and 3 db less than the original —3 for head 3, or —6. Thus the action continues, repetitions at the output always 1 db apart.

So far, the three ganged control potentiometers have always been at maximum. Before moving them, we must explain that they have special tapers, so that in any position the attenuation given by control C is always twice that (in decibels) of controls A or B, which are identical. Thus, if the control is set so that A attenuates its signal 3 db, then control B has 3 db of attenuation and control C attenuates by 6 db.

Let us now reduce the control setting so that A and B have 1 db of attenuation each — and C has 2 db.

The input signal passes through control A and suffers 1 db of attenuation. The output of head 1 is thus 1 db less than with the control wide open. Instead of appearing at the system output at —1, it appears at —2.

The output of head 2 is also 1 db less than before due to control A. It also suffers another 1 db of attenuation in control B, making a total of 2 db. Therefore, instead of being at —2 as originally, it is now at —4.

The output of head 3 is also 1 db less than before due to control A. It also is attenuated 2 db more by control C, making a total of 3. So instead of appearing at the output at —3, it appears as —6.

You will note, therefore, that at this control setting, the system outputs due to the three heads are respectively —2, —4, and —6 with respect to our standard 0-db direct channel. The reverberation time is reduced, but the decay curve, composed of equal decrements in repetitions, is still linear.

Now let us consider the fed-back signal with this control setting. Since the signal to the recording head is attenuated 1 db by control A and control C attenuates the head-3 signal by 2 db more, the fed-back signal arrives at control A 3 db lower than it was before. In going to the recording amplifier it passes through control A, for 1 more decibel of extra attenuation. It therefore comes to the record amplifier attenuated by a total of 4 db more than it did when the control was wide open. When the control was wide open, the fed-back signal produced a system output of —4 db due to head 1. Subtracting 4 db from that, the fed-back signal with the new control setting, due to head 1 is —8 db, which continues the cycle of 2-db decrements as advertised. For a little mental exercise, you can postulate other control settings and additional feedback cycles; you will find that decrements between repetitions are always equal!

Control rotation does not produce linear increments of reverberation time; there is, so to speak, a special "taper."

The numbers on the control dial do not indicate seconds of reverberation time; they are merely reference numbers. As the control is advanced from 0, reverberation time increases rapidly up to a setting of about 1½ or 2 on the dial. As the control is advanced beyond about 2, reverberation time increases much more slowly. The number-2 setting on the dial provides about the minimum reverberation necessary for any effect at all. Thus, virtually unusable reverberation times are relegated to a small portion of the total control rotation while those which are the most useful are spread out over the greatest part of the range.

REVERBATAPE Installation

To add the REVERBATAPE Unit to an audio system of any kind, whether it be an elecronic organ or a high-fidelity, recording, or broadcast system, it is only necessary to find a point in the amplification section where the maximum audio voltage is something between about .3 and 3 volts rms (if the Unit is to be used as normally supplied).

Such a point may be between two voltage-amplifier stages, for instance. Or in a Schober Organ or a high-fidelity system, it may be between the organ or preamplifier output and the power amplifier. The circuit is broken at this point (usually by using existing plugs and sockets or a tube-socket adapter). For example, in a Schober Organ, the organ output, which is normally fed directly to a power amplifier (or to the "auxiliary" position of a high-fidelity system function switch), is fed instead to the input jack of the REVERBATAPE Unit. Then the output jack of the Unit is fed to the amplifier. A survey of the organ field has made it possible to prepare instruction sheets covering many different organs, showing physical placement of the parts, and the connections. In most cases, the necessary cords and connectors are furnished already made up in a small separate Installation Kit tailored for the particular organ.

The Unit itself is attached to any convenient surface within the organ or equipment cabinet with a pair of universal brackets.

Fig. 7 — The REVERBATAPE Control may be mounted on the organ stop board or any cabinet front; a decal is supplied.

The Control may be fastened directly to the organ stop board, as in Fig. 7, or to any wood or metal panel. A plastic control box is also available, for easier installations in some cases. It can be fastened to the organ just below the lower manual (Fig. 8), usually without screws and without marring the woodwork.

In each installation, the input potentiometer must be set in such a way that the maximum available signal will just about fully modulate the recording system. In practice the way to do this, at least with an organ, is to hold down the keys, pedals and stops giving the loudest desired sound and then slowly advance the potentiometer on the printed circuit until obvious distortion sets in; the pot is then

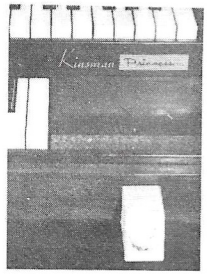
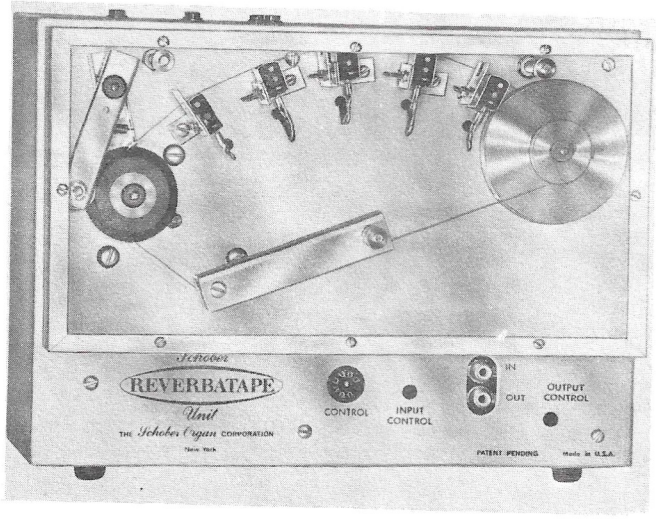


Fig. 8 — A small plastic control box is also available. It will contain the Control and can be mounted below the lower manual of an organ without even requiring screws to hold it in place.

backed off until the distortion just disappears. The output potentiometer may then be set for whatever output is desired — either the same as the input or a different value. Both the input and output potentiometers are available



The Schober REVERBATAPE Unit measures 10½ inches wide, 7⅞ inches high, and 5 inches deep. The clear plastic front cover protects the mechanism.

from the front of the Unit.

Note: The word REVERBATAPE is a registered trademark of The Schober Organ Corporation. Patents have been applied for on the device described in this article.